

WHAT IS CLAIMED IS:

1. A method for approximating $y(n)=1/x(n)$ in FM demodulation, where $x(n)=I^2(n)+Q^2(n)$, comprising:

- (a) receiving a prior estimated value of $1/x(n)$;
- (b) receiving a present value of $x(n)$;
- (c) adjusting the prior estimated value of $1/x(n)$ to compensate for an error between the prior estimated value of $1/x(n)$ and the present value of $1/x(n)$; and
- (d) outputting the adjusted prior estimated value of $1/x(n)$ as the present value of $1/x(n)$.

2. The method of claim 1, wherein the prior estimated value of $1/x(n-1)$ equals $1/(I^2(n-1)+Q^2(n-1))$, wherein $I(n)$ is an input signal and $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$.

3. The method of claim 2, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

4. The method of claim 1, wherein the present value $x(n)$ equals $I^2(n)+Q^2(n)$, and wherein $I(n)$ is an input signal and $Q(n)$ is quadrature-phase signal of $I(n)$.

5. The method of claim 4, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

6. The method of claim 1, wherein an error signal equals $(1-x(n)y(n-1))a$, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1)= 1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and "a" is a scaling coefficient.

7. The method of claim 6, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

8. The method of claim 1, wherein the $Y(n)$ signal equals $y(n-1) + (1-x(n)(y(n-1)))a$, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1) = 1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and "a" is a scaling coefficient.

9. The method of claim 8, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio signal.

10. A method for demodulating an FM signal $FM(n)$ from a secondary audio program signal, comprising:

(a) receiving in-phase $I(n)$ and quadrature-phase $Q(n)$ portions of the $FM(n)$ signal

(b) generating a first portion of the $FM(n)$ signal that is equal to $I(n)Q'(n)-I'(n)Q(n)$;

(c) determining a value $z(n)$ based on the first portion of the $FM(n)$ signal;

(d) generating a second portion of the $FM(n)$ signal that is equal to $1/I^2(n)+Q^2(n)$, wherein $I^2(n)+Q^2(n)$ is equal to $x(n)$ and $y(n)=1/x(n)$;

(e) generating a value for $y(n)$ based on $1/x(n)$ that equals $y(n-1) + (1-x(n)y(n-1))a$; and

(f) multiplying the $z(n)$ value and the $y(n)$ value to produce the $FM(n)$ signal.

11. A system for approximating $y(n)=1/x(n)$ in FM demodulation, where $x(n)=I^2(n)+Q^2(n)$, comprising:

means for receiving a prior estimated value of $1/x(n)$;

means for receiving a present value of $x(n)$;

means for adjusting the prior estimated value of $1/x(n)$ to compensate for an error between the prior estimated value of $1/x(n)$ and the present value of $1/x(n)$; and

means for outputting the adjusted prior estimated value of $1/x(n)$ as the present value of $1/x(n)$.

12. The system of claim 11, wherein the prior estimated value of $1/x(n-1)$ equals $1/(I^2(n-1)+Q^2(n-1))$, wherein $I(n)$ is an input signal and $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$.

13. The system of claim 12, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

14. The system of claim 11, wherein the present value $x(n)$ equals $I^2(n)+Q^2(n)$, and wherein $I(n)$ is an input signal and $Q(n)$ is quadrature-phase signal of $I(n)$.

15. The system of claim 14, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

16. The system of claim 11, wherein an error signal equals $(1-x(n)y(n-1))a$, wherein $x(n)=I^2(n)+Q^2(n)$, $y(n-1)=1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and "a" is a scaling coefficient.

17. The system of claim 16, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

18. The system of claim 11, wherein the $Y(n)$ signal equals $y(n-1) + (1-x(n)(y(n-1)))a$, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1) = 1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and “a” is a scaling coefficient.

19. The system of claim 18, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio signal.

20. A method for approximating $y(n)=1/x(n)$ in FM demodulation, where $x(n)=I^2(n)+Q^2(n)$, comprising:

- (a) receiving $1/x(n-1)$;
- (b) receiving $x(n)$;
- (c) adjusting $1/x(n-1)$ to compensate for an error between $1/x(n-1)$ and $1/x(n)$; and
- (d) outputting the adjusted $1/x(n-1)$ as $1/x(n)$.